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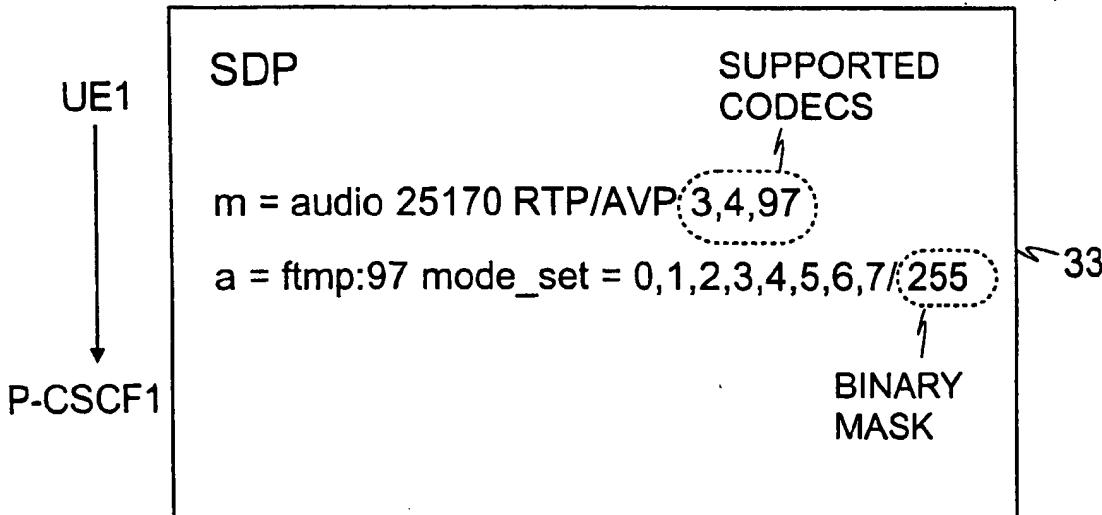
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(54) Title: COMMUNICATION OF CODEC INFORMATION



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(57) Abstract: AbstractThe invention relates to a method for communicating codec related information between a first mobile communication device (UE1) and a second mobile communication device (UE2) via a network. The codec related information relates to operational modes of an adaptive multi-rate (AMR) codec. The information is transmitted by the first mobile communication device (UE1) to the second mobile communication device (UE2) in a form of a binary mask in an SDP (Session Description Protocol) body (33) of a SIP (Session Initiation Protocol) INVITE message (31), a value of 1 indicating the operational modes which are supported by the first mobile communication device (UE1). In the event that the network does not support at least one of the operational modes that the codec related information indicates as being supported the network modifies the codec related information accordingly. Fig. 4a



(BF, BJ, CF, CG, CI, CM, GA, GN, GQ, GW, ML, MR, NE, SN, TD, TG).

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COMMUNICATION OF CODEC INFORMATION

The invention relates to communicating codec related information between a first communication device and a second communication device via a
5 network.

In wireless telecommunication systems information is transferred in an encoded form between a transmitting communication device and a receiving communication device. The transmitting communication device encodes
10 original information into encoded information and sends it to the receiving communication device. The receiving communication device decodes the received encoded information in order to recreate the original information. The encoding and decoding is performed in codecs. Thus, the encoding is performed in a codec located in the transmitting communication device, and
15 the decoding is performed in a codec located in the receiving communication device. However, since there are many different codecs available, the transmitting terminal and the receiving terminal have to agree upon the codec(s) to be used in a session. This agreeing procedure occurs during the initial session establishment and is called a codec negotiation procedure.
20

The codec negotiation procedure for third generation (3G) telecommunication systems is currently being standardised. One of the standard proposals for a codec negotiation procedure for third generation telecommunication systems is discussed in the following with the aid of Figures 1 and 2.
25

Figure 1 shows a third generation telecommunication system for providing codec negotiation. In the system there is formed a signalling chain between a first communication device (hereinafter referred as UE1, UE standing for User Equipment) and a second communication device (hereinafter referred as UE2). The signalling chain goes through a first Proxy Call State Control Function (hereinafter referred as P-CSCF1), a first Serving Call State Control Function (hereinafter referred as S-CSCF1), a second Serving Call State Control Function (hereinafter referred as P-CSCF2), a second Proxy Call
30

- State Control Function (hereinafter referred as S-CSCF2). P-CSCF1, S-CSCF1, P-CSCF2 and S-CSCF2 are logical network entities that may be implemented so as to form separate physical network elements, or they may be incorporated in some of the already existing physical network elements.
- 5 P-CSCF1 and S-CSCF1, for example, may be incorporated in a first GGSN (Gateway General Packet Radio Service (GPRS) Support Node), and they may be controlled by a first network operator. P-CSCF2 and S-CSCF2 may be incorporated in a second GGSN, and they may be controlled by a second network operator. Interfaces between the different devices and functions
- 10 mentioned above are defined in 3GPP (3rd Generation Partnership Project) specifications. It is known to a person skilled in the art that network elements and/or control functions other than the ones shown in Figure 1 may reside in the system.
- 15 The P-CSCF1 and S-CSCF1 are, among other things, responsible for providing services and reserving resources (for example radio resources) for the UE1. The P-CSCF1 controls the UE1 so that it does not exceed the resources that the network is able to provide for it. The S-CSCF1 controls the UE1 so that it does not exceed the resources to which its user has
- 20 subscribed.
- The P-CSCF2 and S-CSCF2 are, among other things, responsible for providing services and reserving resources for the UE2. The P-CSCF2 controls the UE2 so that it does not exceed the resources that the network is able to provide for it. The S-CSCF2 controls the UE2 so that it does not exceed the resources to which its user has subscribed.
- When the UE1 initiates a session with the UE2, the codec to be used for the session is to be determined (negotiated). If the session is going to be a
- 25 multimedia session that is the session is going to be established with more than one media stream (for example an audio stream and a video stream) codecs to be used with each of the streams are to be negotiated.
- 30 .

According to the standard proposal (3G TS 23.228 version 1.7.0) the negotiation is performed in such a way that the UE1 (also referred to as the session originator) first generates, according to the SIP (Session Initiation Protocol) protocol, a SIP INVITE message comprising particular SIP header fields and a message body. According to the proposal, the message body is generated according to the SDP (Session Description Protocol) protocol and it is called an SDP body.

The UE1 generates the SDP body in such a way that it contains a list (set) of codecs that the UE1 is able and willing to support for the session. The UE1 sends the SIP INVITE message to the UE2. When the SIP INVITE message arrives at the UE2, the UE2 responds to the UE1 by generating and sending a reply message, also containing an SDP body, to the UE1. The reply message is referred to in the SIP protocol as the "183 message". The SDP body of the reply message contains a second list of codecs indicating the codecs that the UE2 is able and willing to support for the session. The second list is generated based on the content of the list of codecs in the SDP body of the SIP INVITE message and based on the UE2's ability and willingness to support these codecs. If the UE2 is able and willing to support all the same codecs as the UE1 this results in the second list of codecs being the same as the (original) list of codecs that the UE1 generated in the first place. However, if the UE2 is not able or willing to support, for the session, one or more of the codecs contained in the original list, the UE2 leaves such a codec or such codecs out from the second list. This being the case the second list is a sub-list of the original list. In either case, the second list contains the codecs that both the UE1 and the UE2 are able and willing to support for the session.

When the 183 message, sent by the UE2, arrives at the UE1, the UE1 decides which codec (or codecs if it is a multimedia session) of all of the supported codecs contained in the second list is (or are) to be used in the session. After it has decided this it sends to the UE2 a third message (referred to as the Final SDP) which tells to the UE2 the codec(s) that is (or

are) to be used in the session to be established.

However, if the messages are sent in an end-to-end manner as described above a problem arises, because the decision of the codec(s) to be used is
5 made without determining from the network the capacity that it is able to provide. For example, the chosen codec might be such a codec that requires a larger bandwidth than the network is able to provide at the time in question.

One standard proposal tries to solve this problem by allowing the network
10 entities P-CSCF1, S-CSCF1, S-CSCF2 and P-CSCF2 to remove non-suitable codecs from the codec list in the SDP of the SIP INVITE message. In the following the matter is described in more detail referring now to Figure 2.

After the UE1 has determined the codecs that it supports for the session it
15 sends the SIP INVITE message to the UE2. When the SIP INVITE arrives at the P-CSCF1, on its way to the UE2, the P-CSCF1 removes all non-suitable codec choices from the codec list in the SDP body. By a non-suitable codec choice is meant such a codec in the codec list that is, at the moment (or in general based on a network operator policy), not possible for the session
20 from the network's point of view, the network being the one serving the UE1. One example of a non-suitable codec choice would be a codec that uses too large a bandwidth compared to the bandwidth available in the network.

The P-CSCF1 forwards the message to the S-CSCF1 which removes from
25 the codec list all codecs that the UE1 is not authorised to request (based on user subscription information relating to the user of the UE1).

The S-CSCF1 forwards the message to the S-CSCF2 which removes from
30 the codec list all codecs that the UE2 is not authorised to use (based on user subscription information relating to the user of the UE2).

Also, the S-CSCF1 and S-CSCF2 remove from the codec list all codecs that are not supported based on a network operator policy.

The S-CSCF2 forwards the message to the P-CSCF2 which removes all non-suitable codec choices from the codec list in the SDP body. Again, by a non-suitable codec choice is meant such a codec in the codec list that is, at the 5 moment (or in general based on a network operator policy), not possible for the session from the network's point of view, the network now being the one serving the UE2.

- Finally, the P-CSCF2 forwards the SIP INVITE message to the UE2. The 10 UE2 receives the SIP INVITE message containing the SDP body which now comprises a list of codecs which both the UE1 and all the logical network entities P-CSCF1, S-CSCF1, S-CSCF2 and P-CSCF2 are willing to support for the session.
- 15 The UE2 now responds with a reply message (that is the 183 message) containing a second list of codecs. The second list is generated based on the content of the list of codecs in the SDP body received in the SIP INVITE message and based on the UE2's ability and willingness to support these codecs. If the UE2 is able and willing to support all the codecs contained in 20 the list of codecs, received in the SIP INVITE message, the second list is the same as the list of codecs, received in the SIP INVITE message. If the UE2 is not able or willing to support, for the session, all the codecs contained in the list of codecs, received in the SIP INVITE message, the UE2 leaves such a codec or such codecs out from the second list. In either cases, the second 25 list is a list of codecs that both the UE1 and the UE2 and all the network entities P-CSCF1, S-CSCF1, S-CSCF2 and P-CSCF2 are willing to support for the session.

When the 183 message arrives at the UE1 it can make a choice which 30 automatically takes into account the network capabilities, when deciding the codec(s) to be used initially in the session. Information on the chosen codec is sent to the UE2 in a Final SDP message, in a manner similar to that previously described.

- The method described above only indicates whether a particular codec is supported or not. However, there exist codecs which have multiple options. An AMR (Adaptive Multi Rate) codec, for example, is a codec which supports
- 5 a plurality of different bit rates. In this case it is not enough in the codec negotiation procedure just to indicate whether the AMR codec is supported, but there is a need to indicate the bit rates of the AMR codec that are supported.
- 10 According to a first aspect of the invention there is provided a method for communicating codec related information between a first communication device and a second communication device via a network the method comprising:
transferring from the first communication device to the second
- 15 communication device via the network information indicating, from a group of operational modes of a codec, the operational modes that the first communication device supports for a session between the first communication device and the second communication device, the method comprising:
- 20 modifying the information, by the network, if the information is not acceptable to the network.

The term session is to be construed broadly. The term session shall cover various sessions and connection services in which codecs are to be used.

- 25 Preferably, the network modifies the information if the network does not support at least one of the operational modes which the information indicates as being supported.
- 30 Preferably, the information is received by a network entity in the network, and the network entity modifies the information if the network entity does not support at least one of the operational modes that the information, as received, indicates as being supported.

Preferably, the supported and unsupported operational modes are indicated with the aid of a mask having a plurality of mask elements each mask element being representative of one operational mode.

5

Preferably, each of the plurality of the mask elements indicates whether the corresponding operational mode is supported wherein:

the mask element taking a first value indicates that the operational mode is supported; and

- 10 the mask element taking a second value indicates that the operational mode is unsupported.

- 15 Preferably, the mask is a binary mask and the mask elements are binary digits so that the first value of a mask element is a binary digit 1 and the second value is a binary digit 0.

- 20 Preferably, the information is transferred in a message the message comprising a header portion and a message body the information being transferred in the message body.

20

Preferably, the message body is an SDP (Session Description Protocol) body of a SIP (Session Initiation Protocol) INVITE message.

- 25 Preferably, in the method, it is indicated, from a group of operational modes of a codec, the operational modes that the first communication device is willing and able to support for the session between the first communication device and the second communication device.

- 30 Preferably, at least one of the communication devices is a mobile communication device.

Preferably, the operational modes of the codec are the operational modes/bit rates of an AMR (Adaptive Multi Rate) codec.

- According to a second aspect of the invention there is provided a transmitting communication device for communicating codec related information to a receiving communication device via a network, the transmitting
- 5 communication device comprising:
- a transmitter for transmitting information to the receiving communication device via the network the information indicating, from a group of operational modes of a codec, the operational modes that the transmitting communication device supports for a session between the transmitting
- 10 communication device and the receiving communication device, the transmitting communication device being configured to:
- transmit the information in a format which enables the network to modify the information if the information is not acceptable to the network.
- 15 According to a third aspect of the invention there is provided a system for communicating codec related information between a first communication device and a second communication device the system comprising the first communication device, the second communication device and a network, the first communication device comprising:
- 20 a transmitter for transmitting information from the first communication device to the second communication device via the network the information indicating, from a group of operational modes of a codec, the operational modes that the first communication device supports for a session between the first communication device and the second communication device, the
- 25 network comprising:
- a processing unit for modifying the information if the information is not acceptable to the network.
- According to a fourth aspect of the invention there is provided a message for
- 30 communicating codec related information between a first communication device and a second communication device via a network the message being configured:
- to be transferred from the first communication device to the second

communication device via the network the message having information indicating, from a group of operational modes of a codec, the operational modes that the first communication device supports for a session between the first communication device and the second communication device, the
5 message being configured:
to enable the network to modify the information if the information is not acceptable to the network.

According to a fifth aspect of the invention there is provided a computer
10 program product A computer program product for implementing a network entity the computer program product comprising:
computer executable code for enabling the network entity to handle codec related information being transferred between a first communication device and a second communication device the information indicating, from a group-
15 of operational modes of a codec, the operational modes that the first communication device supports for a session between the first communication device and the second communication device; and
computer executable code for enabling the network entity to modify the information, if the information is not acceptable to the network entity.

20 It is to be understood that the operational modes that are supported may be indicated indirectly. This can be done, for example, in a system (and constituent parts thereof) which uses operational modes from a fixed, predetermined, set of operational modes. If, for example, operational modes
25 that are not supported are indicated, then the supported operational modes should immediately be apparent.

Embodiments of the invention will now be described by way of example only with reference to the accompanying drawings in which:
30 Figure 1 shows a third generation telecommunication system for providing codec negotiation;

- Figure 2 shows a method for codec negotiation in the system presented in Figure 1;
- Figure 3 shows a message structure suitable for codec negotiation;
- 5 Figures 4a to 4e show the content of the message body of a message according to a preferred embodiment of the invention as the message travels through the network;
- 10 Figure 5 shows the content of the message body of another message;
- Figure 6 shows a cellular mobile station suitable for the implementation of the invention; and
- 15 Figure 7 shows a GGSN suitable for the implementation of the invention.
- The system and message sequence shown in Figures 1 and 2 can also be used in a preferred embodiment of the invention. Accordingly, in the preferred embodiment of the invention, a first communication device UE1 first sends to a second communication device UE2 a SIP INVITE message in response to which the UE2 responds with a reply message (for example with a "183 message"). When the UE1 receives the reply message it decides on the codec(s) to be used in a session to be established. The UE1 generates, based on the decision, a third message (Final SDP) and sends the third message containing the information about the decided/chosen codec(s) to the UE2.
- 25
- 30 In the preferred embodiment the UE1 is a wireless mobile station of a cellular radio network and the UE2 is another wireless mobile station of the same or another cellular radio network. An example of the cellular radio network is a wideband code division multiple access (WCDMA) network or another third

generation network.

Figure 3 shows the basic SIP message structure. This is the basic structure of all the three messages sent in the preferred embodiment. A SIP message
5 31 comprises SIP header fields 32 and a message body that is an SDP body
33.

The SIP header fields 32 contain information about the sender and the
recipient of the message such as address information and other general
10 information familiar to a person skilled in the art.

The SDP body 33 contains information concerning those media streams (for
example information on ports and codecs) to be used in a session. Each
media stream is defined in the SDP with the aid of one media line that is an
15 *m-line*. Each media stream may be even more specifically defined with the
aid of one or more attribute lines that is one or more *a-lines* following the *m-*
line.

Let us now assume that the UE1 wants to initiate an audio (speech) session
20 with the UE2. In this exemplary case the UE1 supports the following three
codecs for the audio session: the GSM (Global System for Mobile
communications) codec, the G.723 codec and the AMR codec. The *m-line* for
this media (in the SDP of the SIP INVITE message) would then be like this:

25 *m=audio 25170 RTP/AVP 3,4,97 ,*

wherein *audio* indicates the media type that is audio stream, 25170 indicates
the port number at which the UE1 wants to receive the media, *RTP/AVP*
(Real-Time Transport Protocol/Audio Video Protocol) is the transport protocol
30 to be used and the numbers 3, 4 and 97 indicate the codecs, defined in
RTP/AVP, that the UE1 is willing to support for the session. The mappings
according to *RTP/AVP* are such that number 3 indicates the GSM codec,
number 4 indicates the G.723 codec and number 97 indicates the AMR

codec.

Since the AMR codec has eight different modes of operation so that it can operate with eight different bit rates, these AMR modes/bit rates should also
5 be indicated. According to the preferred embodiment of the invention the bit rates supported are indicated with the aid of an a-line in the SDP body.

The AMR codec itself supports all eight bit rates, but the UE1 might not be able or willing to support all of the bit rates. For example, if UE1 is performing
10 another task simultaneously with the session to be established it may be that the UE1 is not willing to support some of the highest bit rates at the initial stage of the session, although it might, in general, be able to support these bit rates. However, a typical situation is that the UE1 is both able and willing to support all the bit rates.
15

According to the preferred embodiment of the invention the AMR modes/bit rates supported are indicated in the a-line with the aid of a binary mask. The binary mask is a binary number having as many digits as there are different modes/bit rates. Each binary digit corresponds to a mode/bit rate in such a way that each binary digit 1 corresponds to a supported rate and each binary digit 0 corresponds to an unsupported rate. However, in order to consume less space in the SIP messages the binary mask is expressed as a decimal number in the a-line. It is to be noted, that depending on the implementation either the decimal number presentation or the binary number presentation of
20 the binary mask is actually transmitted with the SIP messages.
25

In this exemplary case the UE1 supports all the eight rates. Thus, the a-line (in the SDP of the SIP INVITE message) would look like this:

30 *a=fmtp:97 mode_set=0,1,2,3,4,5,6,7/255 ,*

wherein *fmtp* basically indicates the message body format, 97 indicates that the a-line is for the AMR codec, *mode_set=0,1,2,3,4,5,6,7* indicates the

modes/rates that the AMR codec (of the UE1) itself supports whereas the binary mask 255 indicates which of the bit rates (that is modes) are supported by the device UE1. The meaning of numbers 0 to 7 in the *mode_set* and the use of the binary mask will be explained in greater detail in
5 the following.

In the *mode_set* –list the numbers 0 to 7 correspond the different AMR codec modes/rates in the following way:

10	0	↔	12.2 kbps
	1	↔	10.2 kbps
	2	↔	7.95 kbps
	3	↔	7.40 kbps
	4	↔	6.70 kbps
15	5	↔	5.90 kbps
	6	↔	5.15 kbps
	7	↔	4.75 kbps

If a particular mode number is included in the a-line the corresponding
20 mode/bit rate is supported by the AMR codec. Thus, since all the numbers 0 to 7 appear in the list this is to be construed such that the AMR codec of the UE1 supports all eight modes/rates.

Because the UE1 supports all eight rates the binary mask gets the value
25 255 which corresponds to the decimal number 255. The correspondence between binary digits and AMR codec modes/bit rates is as follows:

255 = 1 1 1 1 1 1 1 1
30 | | | | | | | |
 0 1 2 3 4 5 6 7 (modes/bit rates).

Thus, the mask indicates that all the AMR codec modes/rates 0 to 7 are supported by the UE1, because in the mask there is a binary digit 1
35 corresponding to each of the modes/bit rates.

Now, the UE1 wirelessly sends the SIP INVITE message containing the SDP body comprising the above described *m*-line and *a*-line, as shown in Figure 4a, to the UE2. On its way to the UE2 the network entities P-CSCF1, S-CSCF1, S-CSCF2 and P-CSCF2 remove any non-suitable codec choices
5 from the list of codecs in the *m*-line.

- Relating to the AMR codec, each network entity modifies the mask in the *a*-line if it does not support one or more of the AMR modes/bit rates that the mask indicates as being supported. A network entity might not support a
10 particular AMR codec mode, if for example, the particular AMR mode would use a larger bandwidth than the network is able to provide at the moment. Alternatively, based on a network operator policy, a particular AMR mode, in general, might not be supported.
- 15 The P-CSCF1 receives the SIP INVITE message sent from the UE1. In this exemplary case the P-CSCF1 supports the GSM codec, the G.723 codec and the AMR codec. Relating to the AMR bit rates it supports all bit rates except 12.2 kbps, 7.40 kbps and 5.90 kbps.
- 20 In this exemplary case the P-CSCF1 does not modify the *m*-line, because it supports all codecs that the list contains. However, it has to modify the *a*-line because it does not support some of the bit rates that the mask indicates as being supported. In other words, the P-CSCF1 changes in the binary mask those binary digits corresponding to the unsupported AMR bit rates 12.2 kbps
25 (mode 0), 7.40 kbps (mode 3) and 5.90 kbps (mode 5) from value 1 to value 0. The modification of the mask is illustrated in the following:

255 = 4 1 1 4 1 4 1 1	
↓ ↓ ↓	
30 107 = 0 1 1 0 1 0 1 1	
0 1 2 3 4 5 6 7 (modes/bit rates).	

After the modification the a-line of the SDP body of the SIP INVITE message is:

a=fmtp:97 mode_set=0,1,2,3,4,5,6,7/107.

5

Now, the P-CSCF1 forwards the SIP INVITE message, containing the SDP as illustrated in Figure 4b, to the S-CSCF1. In this exemplary case the S-CSCF1 supports the GSM codec, the G.723 codec and the AMR codec. Relating to the AMR rates it supports all bit rates except 12.2 kbps and 7.95
10 kbps.

In this exemplary case the S-CSCF1 does not modify the *m*-line, because it supports all codecs that the list contains. However, it has to modify the a-line because it does not support some of the AMR modes/bit rates that the mask
15 indicates as being supported.

In other words, the S-CSCF1 changes in the binary mask the binary digit corresponding to the unsupported AMR bit rate 7.95 kbps (mode 2) from value 1 to value 0. Relating to the unsupported bit rate 12.2 kbps (mode 0)
20 the S-CSCF1 does not have to do anything because the binary digit corresponding to that mode/bit rate already has the value 0. The modification of the mask is illustrated in the following:

107 = 0 1 4 0 1 0 1 1
25 ↓
75 = 0 1 0 0 1 0 1 1
| | | | | | | |
0 1 2 3 4 5 6 7 (modes/bit rates).

30 After the modification the a-line of the SDP body of the SIP INVITE message is:

a=fmtp:97 mode_set=0,1,2,3,4,5,6,7/75.

Now, the S-CSCF1 forwards the SIP INVITE message, containing the SDP as illustrated in Figure 4c, to the S-CSCF2. In this exemplary case the S-CSCF2 supports the GSM codec, the G.723 codec and the AMR codec.

5 Relating to the AMR rates it supports all other rates except 12.2 kbps and 7.95 kbps. Thus, the S-CSCF2 does not modify the *m*-line, because it supports all codecs that the list contains, and it does not have to modify the *a*-line either, because the binary digits corresponding to the unsupported AMR bit rates (modes 0 and 2) already have the value 0 (and thus they are

10 not possible AMR bit rates for the session regardless of the capabilities of the S-CSCF2 since if a bit rate is not supported by at least one party transmission at that bit rate is not possible).

Now, the S-CSCF2 forwards the SIP INVITE message to the P-CSCF2. In
15 this exemplary case the P-CSCF2 supports the GSM codec and the AMR codec but it does not support the G.723 codec. Relating to the AMR bit rates it supports all other rates except 12.2 kbps, 6.70 kbps and 5.90 kbps. Now, the P-CSCF2 has to modify both the *m*-line and the *a*-line. It has to modify the *m*-line because it does not support all codecs that the list in the SDP
20 contains. The P-CSCF2 now simply removes from the codec list the unsupported codec choices. In this exemplary case it removes the number 4 which corresponds to the G.723 codec according to the RTP/AVP profile.
After the removal the *m*-line looks like this:

25 *m=audio 25170 RTP/AVP 3,97.*

The P-CSCF2 also has to change the *a*-line because it does not support one of the bit rates that the mask indicates as being supported. The P-CSCF2 changes in the binary mask the binary digit corresponding to the unsupported
30 AMR bit rate 6.70 kbps (mode 4) from value 1 to value 0. Relating to the unsupported bit rates 12.2 kbps (mode 0) and 5.90 kbps (mode 5) the P-CSCF2 does not have to do anything because the binary digits corresponding to these modes/bit rates already have the value 0. The

modification of the mask is illustrated in the following:

75	=	0	1	0	0	4	0	1	1	
↓										
5	67	=	0	1	0	0	0	0	1	1
0 1 2 3 4 5 6 7 (modes/bit rates).										

After the modification the a-line of the SDP body of the SIP INVITE message
10 is:

a=fmtp:97 mode_set=0,1,2,3,4,5,6,7/67.

Finally, the P-CSCF2 forwards the SIP INVITE message containing the SDP
15 body comprising the above modified m-line and a-line, as illustrated in Figure
4d, wirelessly to the UE2.

The UE2 receives the SIP INVITE message. From the content of the SDP
(that is from the m-line and the a-line) of the received SIP INVITE message it
20 is directly derivable which of the codecs both the UE1 and all the network
entities support for the (audio) session. In this case these codecs are the
GSM codec (corresponding to RTP/AVP profile number 3) and the AMR
codec (corresponding to RTP/AVP profile number 97). The *mode_set* in the
a-line tells to the UE2 the modes that the AMR codec itself at the other end
25 supports, that is the modes/rates 0 to 7. The mask in the a-line tells to the
UE2 which of the AMR codec options (modes/bit rates) are supported by
both the UE1 and the network entities. In this case, these are the modes 1
(bit rate 10.2 kbps), 6 (5.15 kbps) and 7 (4.74 kbps).

30 The UE2 now generates the second message which is the reply message (a
183 message or a corresponding message). Again, this is a SIP message
containing an SDP body. The reply message is generated based on the
content of the received SIP INVITE message and based on the UE2's ability

and willingness to support codecs and AMR modes. The reply message also comprises an *m*-line and an *a*-line the content of which is generated based on the content of the *m*-line and *a*-line of the received SIP INVITE message and based on the properties of the UE2. In this exemplary case the port 5 where the UE2 wants to receive the media (that is *audio*) stream is the port number 26250. And the codecs that the UE2 supports for the session are: the GSM codec (number 3) and the AMR codec (number 97). Thus, the *m*-line of the SDP body of the reply message initially looks like this:

10 *m=audio 26250 RTP/AVP 3,97 ,*

wherein 26250 indicates the port number at which the UE2 wants to receive the media, *RTP/AVP* (Real-Time Transport Protocol/Audio Video Protocol) is the transport protocol to be used and the numbers 3 (the GSM codec) and 97 15 (the AMR codec) indicate the codecs, defined in *RTP/AVP*, that the UE2 is willing to support for the session.

The AMR codec of the UE2 supports by definition all the AMR modes/bit rates, and, in this case, also the device UE2 itself, supports all AMR modes. 20 This is a typical case. Thus, the content of the *a*-line of the reply message results in being the same as the *a*-line in the SDP of the SIP INVITE message as received at the UE2, that is:

25 *a=fmtp:97 mode_set=0,1,2,3,4,5,6,7/67 .*

The UE2 sends the reply message, containing the SDP as illustrated in Figure 4e, to the UE1. Although there should be no need for the network entities to modify the binary mask of the reply message further, it may be possible for the network entities to make such a modification if the situation in 30 the network has changed.

When the UE1 receives the reply message it decides, on the basis of the information contained in the reply message, the codec to be used initially in

the (audio) session. In this exemplary case the *m*-line contains two possible codecs: the GSM codec and the AMR codec, and the UE1 decides that the AMR codec is to be used initially in the session.

- 5 The reply message contains also information on possible AMR bit rates. Therefore, it is now possible for the UE1 to decide on the initial AMR source bit rate to be used in the (audio) session. This can be done without any other interaction with the network entities. In this exemplary case the mask in the *a*-line in the reply message indicates three possible modes/bit rates: 10.2
- 10 kbps (mode 1), 5.15 kbps (mode 6) and 4.75 kbps (mode 7), and the UE1 decides that the AMR bit rate 10.2 kbps is to be used initially in the session.

- Now, the UE1 generates the third message (Final SDP or a corresponding message). Again, this is a SIP message containing an SDP body. The UE1
- 15 includes in the SDP body information concerning which codec is to be used initially in the session. If the chosen codec is the AMR codec, like it is in this case, the UE1 also includes in the SDP body information on which AMR bit rate is to be used initially. Other information relating to codecs may be conveyed in the third message, for example additional information about
 - 20 other bit rates and other codecs that may be used. Thus, if the codec and/or bit rate has to be changed during the established session the possible choices would already be known to the UE1 and the UE2.

In the following, some other embodiments of the invention will be introduced.

25

- In the preferred embodiment of the invention, described in the foregoing, it is assumed that the AMR bit rate which is actually used for transmission is the same for both directions that is from UE1 to UE2 and vice versa. In one alternative embodiment it is assumed that the communication devices UE1
- 30 and UE2 may use different AMR bit rates in different directions. According to this embodiment the UE1 includes in the SDP of the SIP INVITE message two masks 51 and 52 (Figure 5). One mask (mask 51) may be modified by the network entities concerning the supported AMR bit rates in one direction

(from the UE1 to the UE2). The other mask (mask 52) may be modified by the network entities concerning the supported AMR bit rates in the other direction (from the UE2 to the UE1).

- 5 According to another embodiment of the invention the UE1 includes in the SDP of the SIP INVITE message even more masks, one for each network entity. Each network entity may now modify its own mask, when the SIP INVITE message passes through the network entity, indicating supported AMR bit rates by binary digit 1 and unsupported AMR bit rates by binary digit 0. In this way the communication devices UE1 and UE2 are able to determine the bit rates which the individual network entities support.
- 10

According to yet another embodiment of the invention the binary mask in the SIP INVITE message is presented as a hexadecimal number instead of the decimal number. Since hexadecimal numbers, in general, require less space than the corresponding decimal numbers, this leads to savings in the required space in the message. For example, the binary mask 01101011 is 107 when presented as a decimal number and is 6B when presented as a hexadecimal number.

- 20
- If it is assumed that the modes/bit rates that the AMR codec supports are generally known, it is possible to omit the numbers indicating the modes/bit rates from the *mode_set*-list of the a-line. According to an embodiment of the invention the *mode_set* -list of the messages contains only the binary mask without the numbers of the modes/bit rates as follows:
- 25

a=fmtp:97 mode_set=67 ,

wherein the decimal number 67 represents the modifiable binary mask. In
30 this way it is possible to save space in the messages.

The invention may be implemented by software. Figure 6 shows a cellular mobile station 60 suitable for the implementing the invention. The mobile

station 60 shown operates as the UE1. A corresponding mobile station may operate as the UE2. The mobile station 60 comprises a processing unit CPU, a radio frequency part RF and a user interface UI. The radio frequency part RF and the user interface UI are coupled to the processing unit CPU. The 5 user interface UI comprises a display and a keyboard (not shown) to enable a user to use the mobile station 60. In addition, the user interface UI comprises a microphone and a speaker for receiving and producing audio signals. The processing unit CPU comprises a microprocessor (not shown), memory MEM and software SW. The software SW is stored in the memory 10 MEM. The microprocessor controls, on the basis of the software SW, the operation of the mobile station 60, such as the use of the radio frequency part RF and the presenting of information in the user interface UI and the reading of inputs received from the user interface UI. The software SW comprises a WCDMA protocol stack on the basis of which a transmitter (not 15 shown) of the radio frequency part RF transmits and a receiver (not shown) of the radio frequency part RF receives messages and other information with the aid of its antenna ANT. The codecs the support of which is negotiated reside in the mobile station 60. They may be implemented in the software SW. Another alternative is hardware implementation of the codecs (not 20 shown).

Figure 7 shows a GGSN suitable for implementing the invention. The GGSN shown serves the UE1 and a corresponding one serves the UE2. The GGSNs may be controlled by different network operators. The GGSN 25 comprises a cellular network interface 71, a control unit 72 and a GGSN interface 73. The cellular network interface 71 and the GGSN interface 73 are coupled to the control unit 72. The GGSN sends and receives information to and from the UE1 via the cellular network interface 71. Typically, there are several other network elements between the GGSN and the UE1. These 30 network elements such as a base station, a base station controller and a SGSN (Serving GPRS Support Node) are well known by a person skilled in the art. The GGSN sends and receives information to and from the GGSN serving the UE2 via the GGSN interface 73. The latter GGSN then has a

corresponding cellular network interface for communicating information with the UE2.

- The network entities P-CSCF1, S-CSCF1, S-CSCF2 and P-CSCF2 are
5 logical network entities implemented by software. The network entities may
be implemented so as to form separate physical network elements, or they
may be incorporated in some of the already existing physical network
elements. In this embodiment, the network entities P-CSCF1 and S-CSCF1
are incorporated in a first GGSN and coupled with the control unit of that
10 GGSN, and the network entities S-CSCF2 and P-CSCF2 are incorporated in
a second GGSN and coupled with the control unit of that GGSN.
Alternatively, the logical network entities can be located in another computer,
but are linked with the GGSN.
- 15 The control unit 72 comprises a processor or another processing unit,
memory and software comprising a program code. The software is stored in
the memory. The processor controls, on the basis of the software, the
operation of the GGSN, such as the use of the cellular network interface 71
and the GGSN interface 73. The processor of the first GGSN implements the
20 functionality of the logical network entities P-CSCF1 and S-CSCF1, and the
processor of the second GGSN implements the functionality of the logical
network entities S-CSCF2 and P-CSCF2.

- As to the method according to the invention, the microprocessor of the
25 mobile station UE1 (Fig. 6) generates the SIP INVITE message containing
the binary mask, by using the software SW. It forwards the SIP INVITE
message to the radio frequency part RF which transmits the SIP INVITE
message wirelessly to the base station of a cellular network from which the
message is conveyed to the first GGSN (serving the UE1). The first GGSN
30 receives the SIP INVITE message via the cellular network interface 71. The
processor of the control unit 72 implements the modification of the *m*-line
(removal of unsupported codecs) and the modification of the *a*-line
(modification of the binary mask) of the SDP body according to the logical

network entity P-CSCF1. Thereafter the processor of the control unit 72 implements the modification of the *m*-line (removal of unsupported codecs) and the modification of the *a*-line (modification of the binary mask) of the SDP body according to the logical network entity S-CSCF1. Here it is to be
5 understood that although the preferred embodiment of the invention talks about forwarding the SIP INVITE message from the P-CSCF1 to the S-CSCF1 the forwarding of the message may occur, instead of a physical forwarding, by another type of forwarding where the message content just is transferred from one software process to another in one and the same
10 device/computer.

The control unit 72 uses the GGSN interface 73 in forwarding the SIP INVITE message to the second GGSN (the one serving the UE2). The second GGSN receives the SIP INVITE message via the GGSN interface 73. The processor
15 of the control unit 72 implements the modification of the *m*-line (removal of unsupported codecs) and the modification of the *a*-line (modification of the binary mask) of the SDP body according to the logical network entity S-CSCF2. Thereafter the processor of the control unit 72 implements the modification of the *m*-line (removal of unsupported codecs) and the
20 modification of the *a*-line (modification of the binary mask) of the SDP body according to the logical network entity P-CSCF2. Thereafter the second GGSN forwards the SIP INVITE message to the UE2 via the cellular network interface 71.

25 The radio frequency part RF of the UE2 receives the SIP INVITE message via its antenna ANT (Fig. 6) and forwards the SIP INVITE message to the processing unit CPU. The microprocessor of the processing unit CPU handles the SIP INVITE message and generates the reply message. It sends the reply message via the two GGSNs to the UE1. The microprocessor of the
30 UE1 decides the codec(s) and bit rate(s) to be used initially in the session. It generates the third message and transmits it to the UE2 via the two GGSNs.

According to the invention, it is possible to provide for the communication

devices UE1 and UE2 information about network capabilities. It is possible to define which of the suggested codecs are supported and which are not supported by the network. If a particular/single codec has multiple choices/options, it is also possible to provide information about whether these 5 choices/options are supported. It is, for example, possible to tell to the communication devices UE1 and UE2 which AMR modes/source bit rates are supported by the network. When using the binary mask indicating the supported and unsupported modes/bit rates it is possible to consume less space in the messages relating to codec negotiation.

10

In addition to the codec negotiation procedure presented in the preceding description the basic message structure and the use of the mask is also applicable in other codec negotiation procedures where the message sequence may deviate from the one presented. It is clear to a person skilled 15 in the art that the messages also contain other information, than the *m*- and *a*-lines, this other information being apparent to a skilled person but not shown in the Figures. If it is evident that the UE1 supports all the same AMR modes/bit rates as the AMR codec itself, the UE1 does not necessarily have to insert the mask to the SIP INVITE message, but the mask may be inserted 20 by the first network entity that does not support at least one of the modes/bit rates, for example the P-CSCF1. It is also possible to use some other message body than the SDP body in the messages. In this case, lines other than the *m*-lines and *a*-lines may be used, although the principle of the binary mask is still applicable. The invention is not restricted to the particular names 25 of the messages (SIP INVITE, 183 message and FINAL SDP).

Particular implementations and embodiments of the invention have been described. It is clear to a person skilled in the art that the invention is not restricted to details of the embodiments presented above, but that it can be 30 implemented in other embodiments using equivalent means without deviating from the characteristics of the invention. The scope of the invention is only restricted by the attached patent claims.

Claims

1. A method for communicating codec related information between a first communication device (UE1) and a second communication device (UE2) via a network the method comprising:
 - 5 transferring from the first communication device (UE1) to the second communication device (UE2) via the network information indicating, from a group of operational modes of a codec, the operational modes that the first communication device supports for a session between the first communication device and the second communication device, **characterised by**
 - 10 modifying the information, by the network, if the information is not acceptable to the network.
- 15 2. A method according to claim 1, wherein the network modifies the information if the network does not support at least one of the operational modes which the information indicates as being supported.
3. A method according to claim 1 or claim 2, wherein the information is received by a network entity in the network, and the network entity modifies the information if the network entity does not support at least 20 one of the operational modes that the information, as received, indicates as being supported.
- 25 4. A method according to any of the preceding claims, wherein the supported and unsupported operational modes are indicated with the aid of a mask having a plurality of mask elements each mask element being representative of one operational mode.
- 30 5. A method according to claim 4, wherein each of the plurality of the mask elements indicates whether the corresponding operational mode is supported wherein:
 - the mask element taking a first value indicates that the

operational mode is supported; and

the mask element taking a second value indicates that the operational mode is unsupported.

- 5 6. A method according to claim 5, wherein the mask is a binary mask and
the mask elements are binary digits so that the first value of a mask
element is a binary digit 1 and the second value is a binary digit 0.
- 10 7. A method according to any of the preceding claims, wherein the information is transferred in a message (31) transmitted from the first communication device to the second communication device via the network.
- 15 8. A method according to claim 7; wherein the message (31) comprises a header portion (32) and a message body (33) and the information is transferred in the message body (33).
- 20 9. A method according to claim 8, wherein the message body is an SDP (Session Description Protocol) body of a SIP (Session Initiation Protocol) INVITE message.
10. A method according to any of the preceding claims, wherein the method comprises:
25 indicating, from a group of operational modes of a codec, the operational modes that the first communication device (UE1) is willing and able to support for the session between the first communication device (UE1) and the second communication device (UE2).
- 30 11. A method according to any of the preceding claims, wherein at least one of the communication devices (UE1, UE2) is a mobile communication device.
12. A method according to any of the preceding claims, wherein the

operational modes of the codec are the operational modes/bit rates of an AMR (Adaptive Multi Rate) codec.

13. A transmitting communication device (UE1) for communicating codec related information to a receiving communication device (UE2) via a network, the transmitting communication device comprising:
 - a transmitter (RF) for transmitting information to the receiving communication device (UE2) via the network the information indicating, from a group of operational modes of a codec, the operational modes that the transmitting communication device supports for a session between the transmitting communication device and the receiving communication device, **characterised** in that the transmitting communication device (UE1) is configured to:
 - transmit the information in a format which enables the network to modify the information if the information is not acceptable to the network.
14. A system for communicating codec related information between a first communication device (UE1) and a second communication device (UE2) the system comprising the first communication device, the second communication device and a network, the first communication device (UE1) comprising:
 - a transmitter (RF) for transmitting information from the first communication device to the second communication device (UE2) via the network the information indicating, from a group of operational modes of a codec, the operational modes that the first communication device (UE1) supports for a session between the first communication device and the second communication device, **characterised** in that the network comprises:
 - a processing unit (72) for modifying the information if the information is not acceptable to the network.
15. A message (31) for communicating codec related information between

a first communication device (UE1) and a second communication device (UE2) via a network the message being configured:

to be transferred from the first communication device (UE1) to the second communication device via the network the message (31)

5 having information indicating, from a group of operational modes of a codec, the operational modes that the first communication device supports for a session between the first communication device and the second communication device, **characterised in that the message (31) is configured:**

10 to enable the network to modify the information if the information is not acceptable to the network.

16. A computer program product for implementing a network entity (P-CSCF1, P-CSCF2, S-CSCF1, S-CSCF2) the computer program product comprising:

computer executable code for enabling the network entity to handle codec related information being transferred between a first communication device (UE1) and a second communication device (UE2) the information indicating, from a group of operational modes of a codec, the operational modes that the first communication device supports for a session between the first communication device and the second communication device; and

25 computer executable code for enabling the network entity (P-CSCF1, P-CSCF2, S-CSCF1, S-CSCF2) to modify the information, if the information is not acceptable to the network entity.

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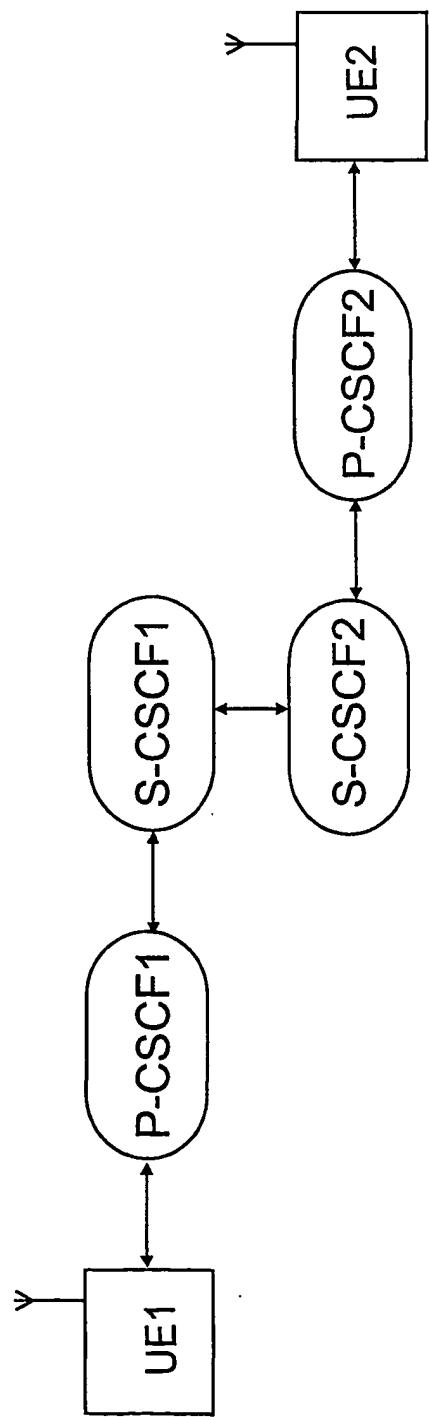


Fig. 1

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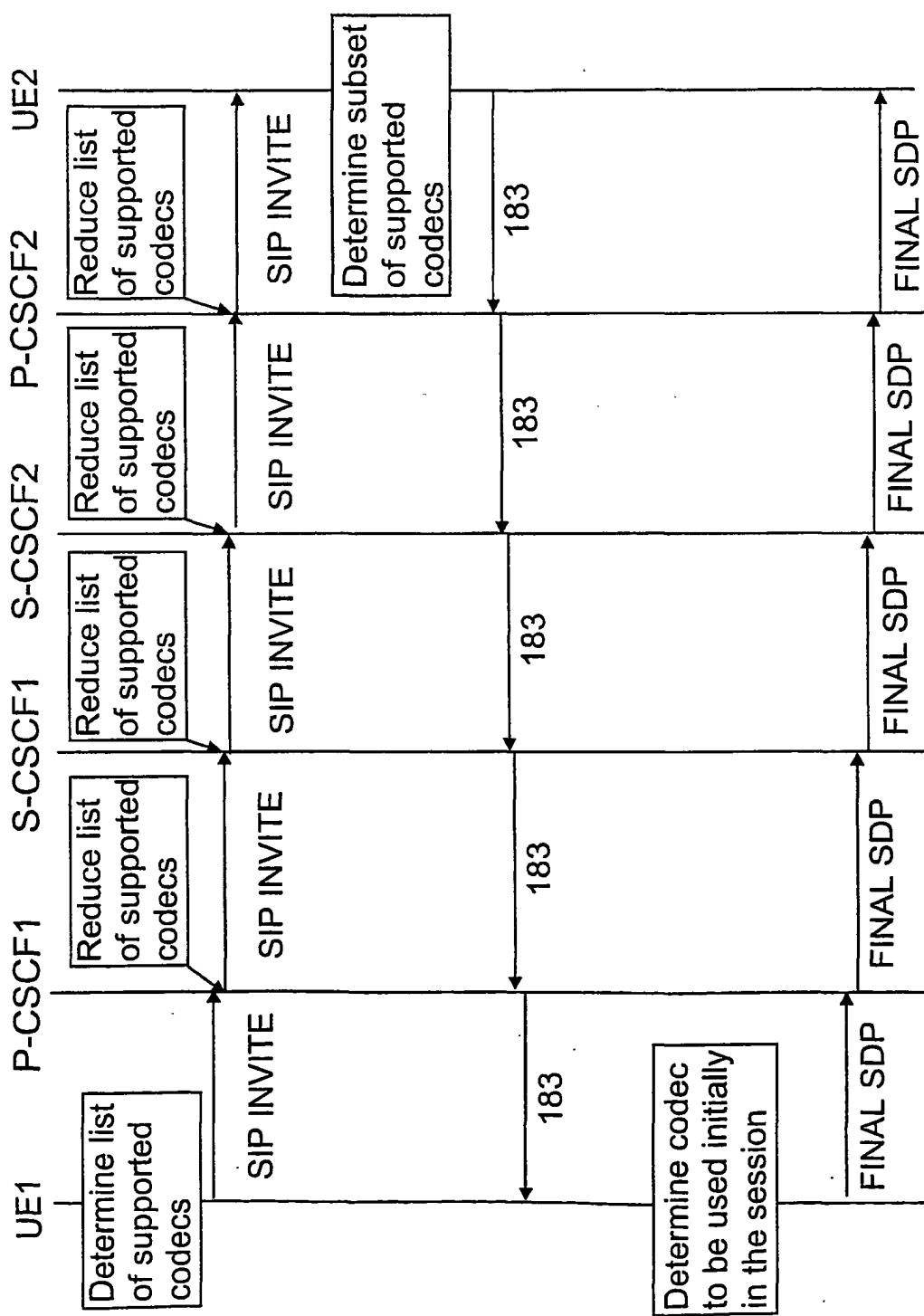


Fig. 2

3/7

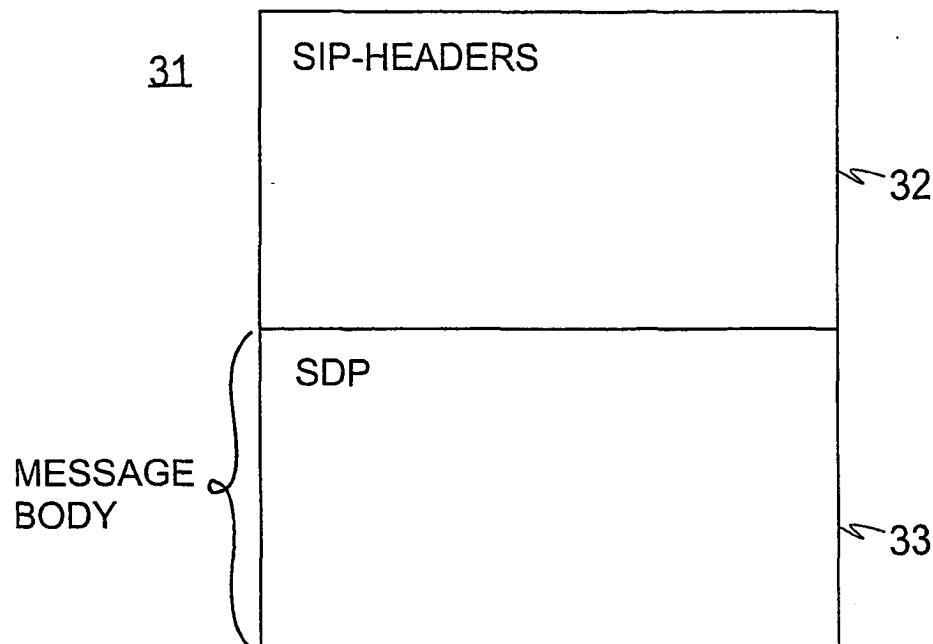


Fig. 3

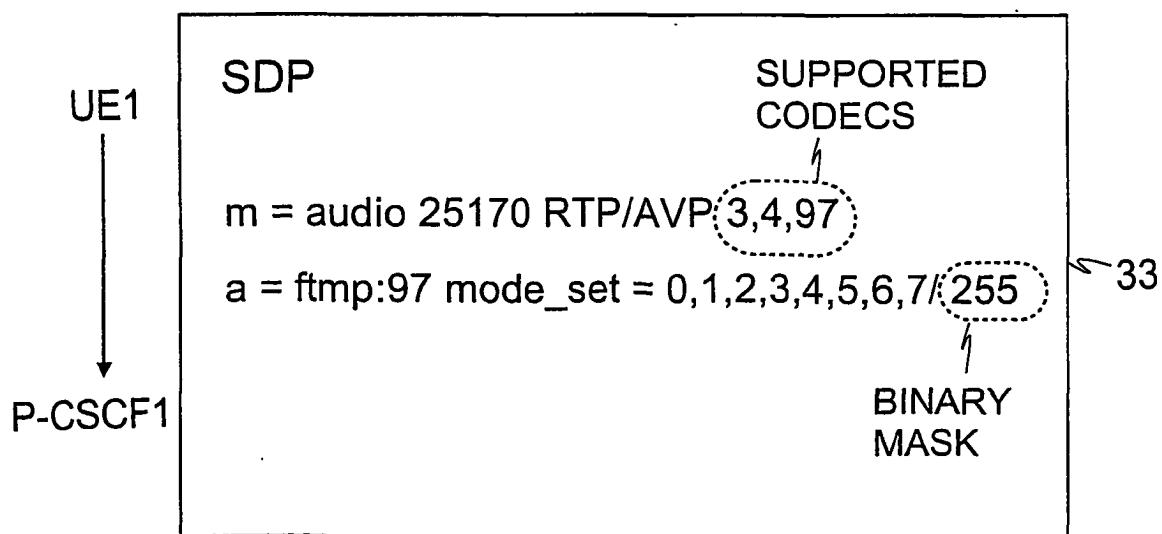


Fig. 4a

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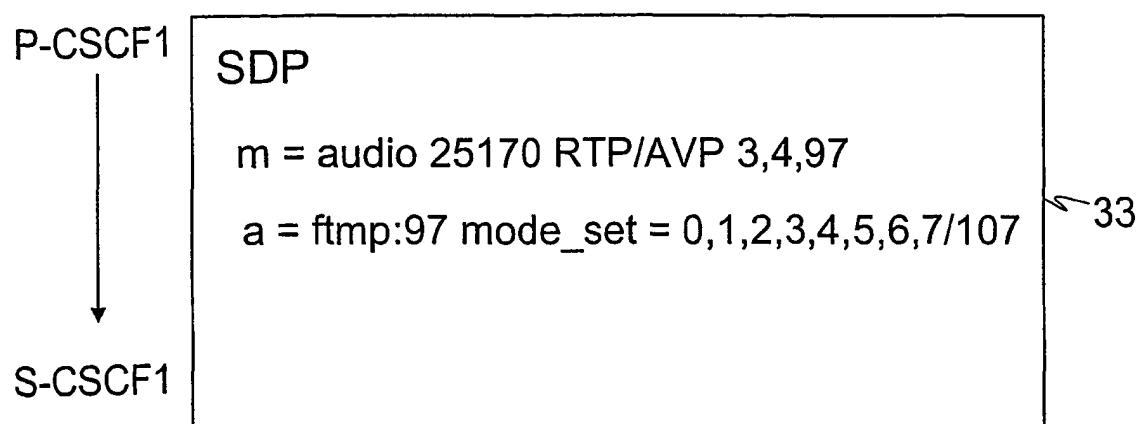


Fig. 4b

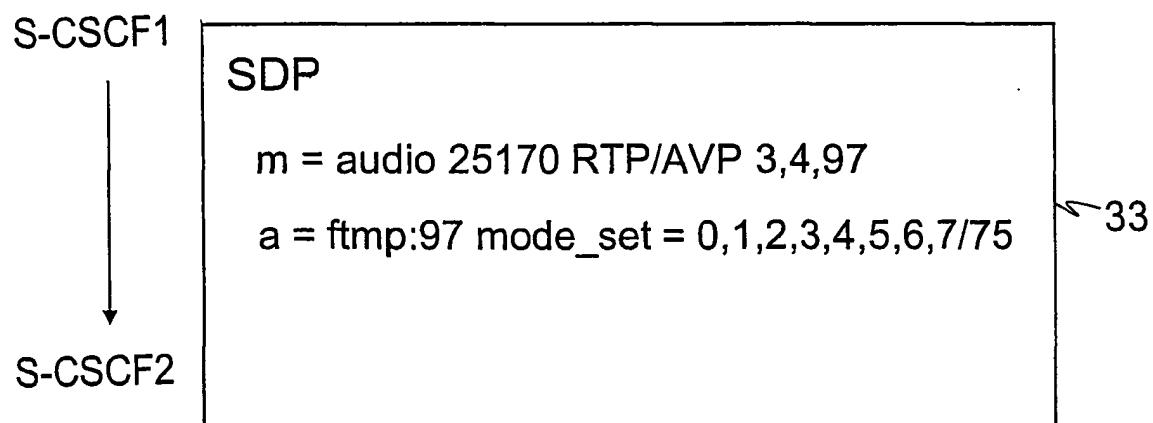


Fig. 4c

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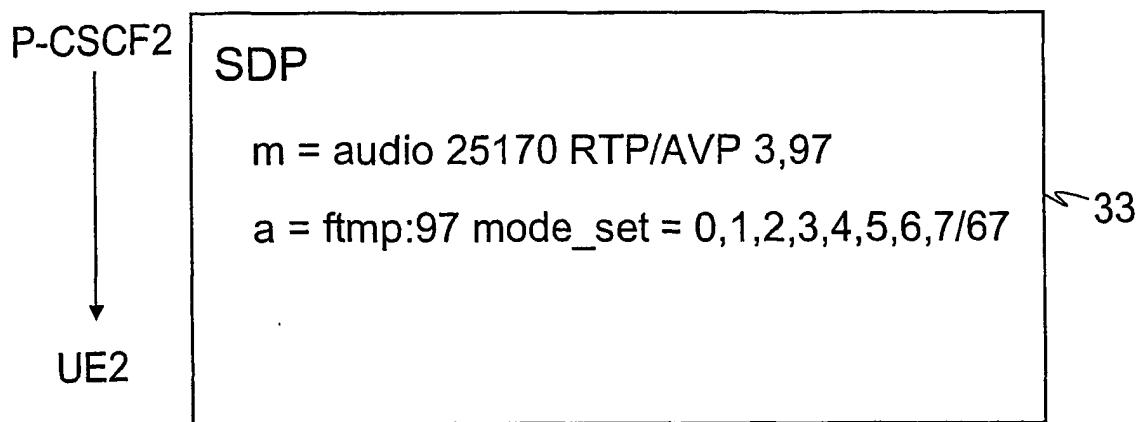


Fig. 4d

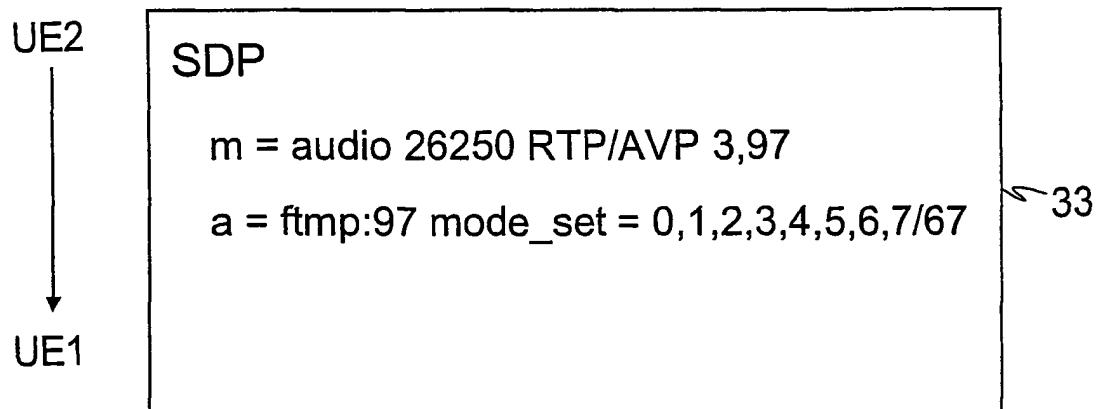


Fig. 4e

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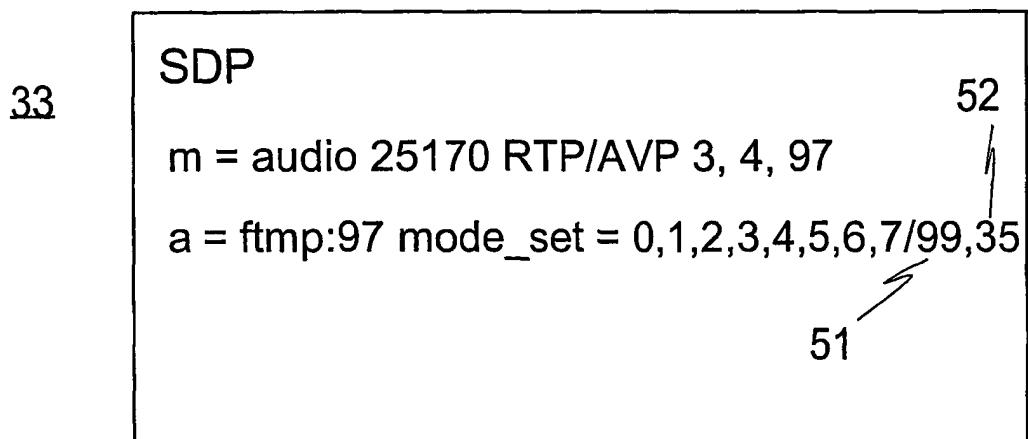


Fig. 5

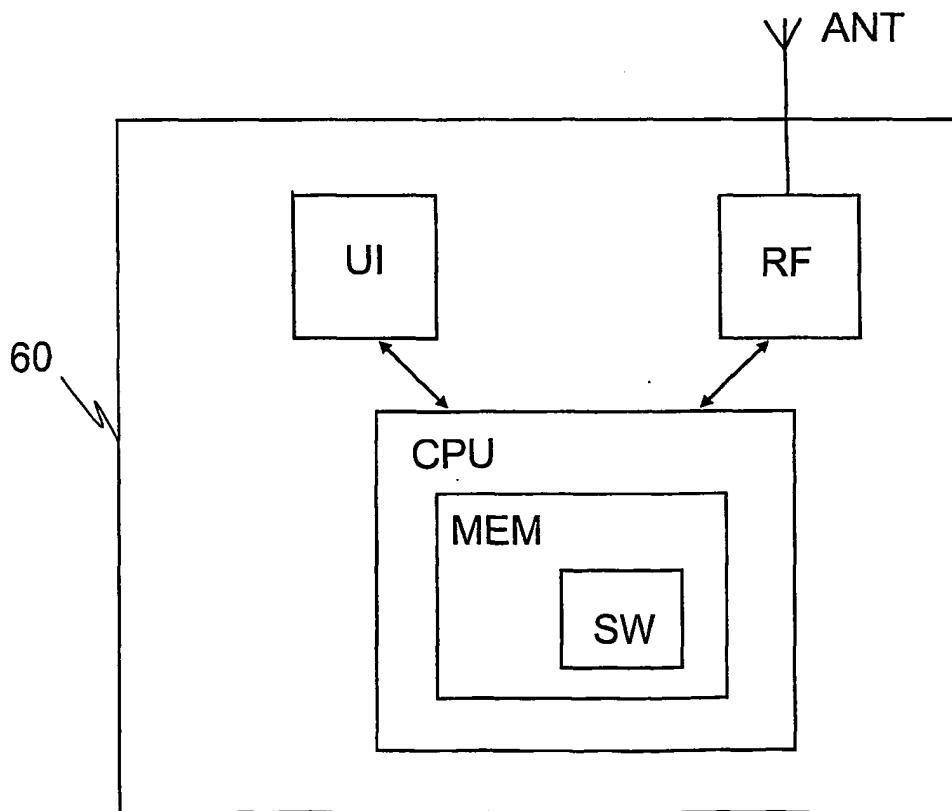


Fig. 6

7/7

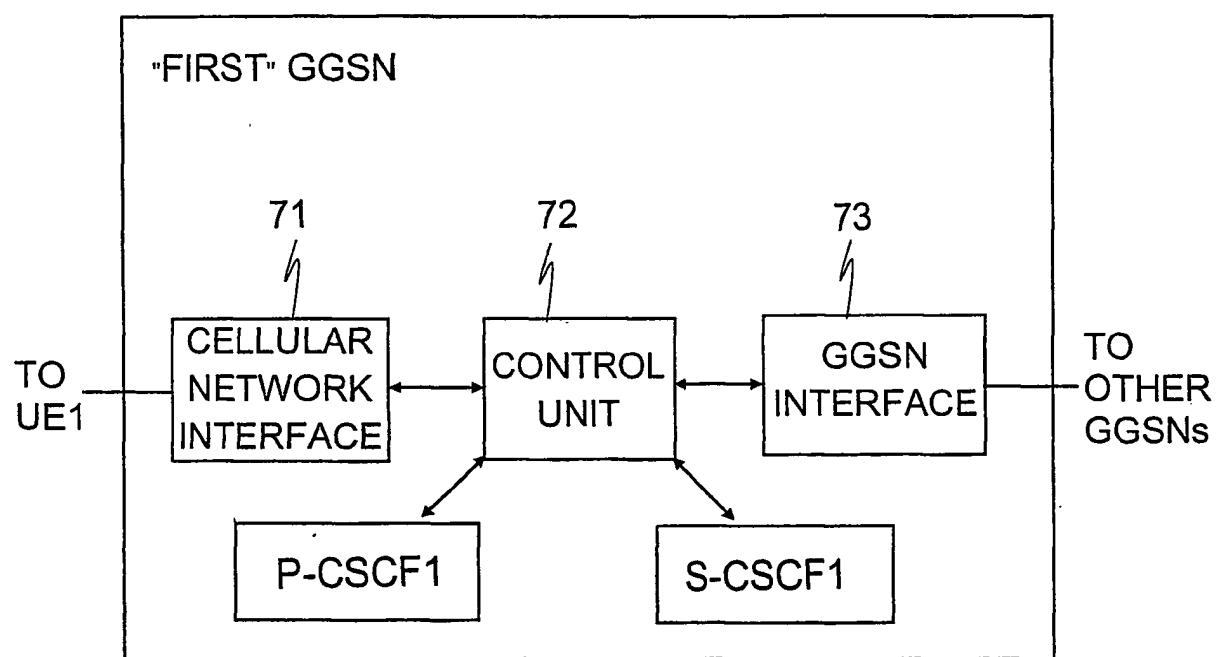


Fig. 7

INTERNATIONAL SEARCH REPORT

International application No.

PCT/FI 02/00434

A. CLASSIFICATION OF SUBJECT MATTER

IPC7: H04Q 7/38, H04L 12/58

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC7: H04L, H04Q

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

SE,DK,FI,NO classes as above

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

EPO-INTERNAL, WPI DATA, PAJ, INSPEC

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	WO 0070885 A1 (TELEFONAKTIEBOLAGET LM ERICSSON (PUBL)), 23 November 2000 (23.11.00), page 3, line 6 - page 8, line 23 --	1-16
P,X	WO 0215625 A1 (NOKIA CORPORATION), 21 February 2002 (21.02.02), page 2, line 31 - page 6, line 23 --	1-16
A	US 6108560 A (WILLIAM NAVARO ET AL), 22 August 2000 (22.08.00), column 2, line 47 - column 4, line 6 --	1-16

 Further documents are listed in the continuation of Box C. See patent family annex.

* Special categories of cited documents:	
"A"	document defining the general state of the art which is not considered to be of particular relevance
"E"	earlier application or patent but published on or after the international filing date
"L"	document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)
"O"	document referring to an oral disclosure, use, exhibition or other means
"P"	document published prior to the international filing date but later than the priority date claimed
"T"	later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"X"	document of particular relevance: the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"Y"	document of particular relevance: the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art
"&"	document member of the same patent family

Date of the actual completion of the international search

20 Sept 2002

Date of mailing of the international search report

26-09-2002

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INTERNATIONAL SEARCH REPORT

International application No.
PCT/FI 02/00434

C (Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
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A	EP 0964540 A2 (TEXAS INSTRUMENTS INCORPORATED DALLAS), 15 December 1999 (15.12.99), page 3, line 45 - page 5, line 21 -----	1-16

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Information on patent family members

02/09/02

International application No.

PCT/FI 02/00434

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		GB	2350261 A	22/11/00
		GB	9911441 D	00/00/00
		GB	9921647 D	00/00/00
		GB	2351414 A	27/12/00
		GB	9914654 D	00/00/00
		GB	2351416 A	27/12/00
		GB	9914700 D	00/00/00
		GB	2352127 A	17/01/01
		GB	9915366 D	00/00/00

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